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L14: Entry 4 of 9 File: USPT Feb 29, 2000

US-PAT-NO: 6031818
DOCUMENT-IDENTIFIER: US 6031818 A

TITLE: Error correction system for packet switching networks

DATE-ISSUED: February 29, 2000

INVENTOR-INFORMATION:

NAME	CITY	STATE	ZIP CODE	COUNTRY
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Padmanabhan; Krishnan	New Providence	NJ		

US-CL-CURRENT: 370/216; 370/390, 370/394, 709/203, 714/748

CLAIMS:

What is claimed:

1. A data packet retransmission arrangement comprising:

at least one client unit associated with at least one destination;

a source of data packets, said source transmitting the data packets to said at least one client unit using a real-time transport protocol, said data packets representing real-time audio or video, said client unit operable for receiving said data packets on behalf of said at least one destination; and

at least one server unit operable to receive a copy of each said data packet that said source transmits to said client unit;

said client unit being further operable, responsive to a received packet being indicative of an error, for sending to said server, rather than to said source, a request to retransmit that one of the data packets that said server received from said source that will correct said error;

wherein said at least one client unit and said at least one server unit operate independently of said source and said at least one destination.

2. The arrangement according to claim 1, wherein said client unit includes:

a detector for determining if an incoming data packet is in a proper sequence;

a playback buffer for storing each said incoming data packet in a time order; and

means for generating said request to retransmit to said server unit if said incoming data packet is out of sequence.

3. The arrangement according to claim 2, wherein said detector calculates an inter-packet delay if said incoming data packet is out of sequence and inserts each said out of sequence incoming data packet in said playback buffer to form a repaired packet stream.

4. The arrangement according to claim 3, wherein a repaired packet stream is sent from said playback buffer to said at least one destination in said time order.

5. The arrangement according to claim 1, wherein said client unit sends out at least another request to retransmit after a given interval.

6. The arrangement according to claim 5, wherein said given interval is a multiple of an average round trip time between said client unit and said server unit.

7. The arrangement according to claim 1, wherein said server unit responds to said request to retransmit by retransmitting a lost data packet to said client unit.

8. The arrangement according to claim 7, wherein said request to retransmit includes a sequence number, a retransmission request number and a requesting client unit number.

9. The arrangement according to claim 7, wherein said server unit includes a retransmit buffer for storing a given number of data packets.

10. The arrangement according to claim 1, wherein said data packets are sent using a real time transport protocol, further wherein:

said client unit is operable to determine if an incoming data packet is out of sequence and is further operable to generate said request to retransmit for a lost data packet if said incoming data packet is out of sequence;

said server unit being responsive to said request by sending said copy of said lost data packet to said client unit; and

said client unit being further operable to send said lost data packet in a time order to said destination.

11. A method for retransmitting data packets, said method comprising the steps of:

transmitting data packets from a source to at least one client and to at least one server using a real-time transport protocol, said data packets representing real-time audio or video, said client unit associated with at least one destination and operable for receiving said data packets on behalf of said at least one destination; and

responding to a received packet being indicative of an error, by having said client request retransmission of said data packets from said server, rather than said source, that will correct said error;

wherein said at least one client and said at least one server operate independently of said source and said at least one destination.

12. The method according to claim 11, wherein said step of responding includes the step of correcting said error in said client unit by forming a repaired packet stream.

13. The method according to claim 11, wherein said step of responding further includes the steps of:

determining if an incoming data packet is in a proper sequence;

buffering a given set of said data packets in said client unit in a given time order in response to whether said incoming data packet is out of sequence;

generating a retransmission request with a set of identification parameters if said incoming data packet is out of sequence and is a lost data packet;

transporting a copy of said lost data packet from said server unit to said client unit; and

inserting said copy of said lost data packet in said time order to form a repaired packet stream.

14. The method according to claim 11, wherein said step of responding includes the step of sending another retransmission request after waiting a given duration for a copy of a lost data packet from said server unit.

15. The method according to claim 11, further including the step of generating at least one lost data packet indicator corresponding to each said error.

16. An apparatus for retransmitting data packets, said apparatus comprising:

a playback register associated with at least one destination;

a source of data packets, said source transmitting the data packets to said playback register using a real-time transport protocol, said data packets representing real-time audio or video, said playback register operable for receiving said data packets on behalf of said at least one destination; and

a retransmit register for receiving a copy of each said data packets that said source transmits to said playback register;

said playback register being further operable, responsive to a received packet indicative of an error, for sending to said retransmit buffer, rather than to said source, a request to retransmit that one of the data packets that said retransmit register received from said source that will correct said error;

wherein said playback register and said retransmit register operate independently of said source and said at least one destination.

17. The apparatus according to claim 16, wherein:

said playback register is operable to determine an out of sequence incoming packet and hold a given set of packets;

said retransmit register is operable to send a copy of a lost data packet to said playback register; and

said playback register is further operable to insert said copy of said lost data packet from said retransmit register into said set of packets in a time order to form a repaired packet stream.

18. The apparatus according to claim 16, wherein said data packets are sent over a real time transport protocol and said playback register uses said protocol format to place a set of identifying information regarding a lost packet.

19. The apparatus of claim 16, wherein said playback register is a circular register and has an active region.

20. The apparatus of claim 16, wherein said playback register maintains a time order of said data packets and transmits said data packets at a given inter-packet interval.

21. A system for transmission of packet streams between at least one source and at least one receiver using a real-time transport protocol, said data packets representing real-time audio or video, said system comprising:

at least one server unit being operable to receive a packet stream from said source; and

at least one client unit being operable to detect an error in said packet stream from said source and request retransmission of a lost packet from said server, rather than from said source;

said client unit being further operable to send a repaired packet stream to said receiver, said repaired packet stream including said lost packet;

wherein said at least one client unit and said at least one server unit operate independently of said source and said receiver.

22. The system according to claim 21, wherein:

said client unit being further operable to determine if a packet is out of sequence; and

said client unit being further operable to store said packet and mark an intermediate set of packets as lost if said packet is out of sequence.

23. The system according to claim 21, wherein:

said client unit being operable to insert a retransmitted lost packet in a time order to form said repaired packet stream; and

said client unit being further operable to send said repaired packet stream in said time order to said receiver.

24. The system according to claim 21, wherein:

said server unit being further operable to store M number of packets, each said packet having a sequence number; and

said server unit being further operable to retrieve said packet by locating said packet by said sequence number modulo M.

25. The system according to claim 24, wherein said M is a power of 2.

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L14: Entry 5 of 9 File: USPT Oct 5, 1999

US-PAT-NO: 5963551
DOCUMENT-IDENTIFIER: US 5963551 A

TITLE: System and method for dynamically reconfigurable packet transmission

DATE-ISSUED: October 5, 1999

INVENTOR-INFORMATION:

NAME	CITY	STATE	ZIP CODE	COUNTRY
Minko; Jacek	San Jose	CA		

US-CL-CURRENT: 370/356; 370/394, 370/477

CLAIMS:

What is claimed is:

1. A system for communicating digital audio signals over a randomly connected link between two transceivers, comprising:

first and second sender/receiver units communicating with each other via a random network connection and configured for both transmitting and receiving data packets representative of audio information, each packet containing an index to uniquely identify the packet;

means for causing the first sender/receiver unit to start a counter and an adjustment timer to determine a number of lost packets for a specified time period, if a difference between the index of a new packet and the index of a last previous packet is greater than one; and

means for prompting the second sender/receiver unit to retransmit to the first sender/receiver unit at least some of the data packets when the number of lost packets for the specified time exceeds a predetermined threshold.

2. A method for recovering lost packets representing audio data in a network environment comprising steps, performed by a receiving station in the network, of:

receiving a new audio data packet from a sending station in the network, wherein the new packet contains a packet index, wherein the packet index is a numerical value that identifies the order of transmission of the new packet;

storing a packet index of a last previous packet in a memory;

comparing the stored packet index and the packet index of the new packet to yield a difference;

if the difference is greater than one, starting an adjustment timer and a the counter to determine a number of lost audio packets for a specific period of time;

initiating a lost audio packet recovery procedure, if the adjustment timer and the counter show that the number of lost packets counted by the counter exceeds a predetermined threshold for the number of lost audio packets for the time specified by the adjustment timer; and

forwarding lost audio packet recovery information to the sending station, as part of

the lost audio packet recovery procedures.

3. The method as recited in claim 2, further comprising the step of:

if the difference exceeds one half the span of possible packet index values, considering the newly received audio packet out-of-sequence and discarding the newly received packet.

4. The method as recited in claim 2, further comprising the step of:

if the adjustment timer reaches the specified time and the count is under a predetermined threshold for a predetermined period of time, terminating the lost audio packet recovery process.

5. The method as recited in claim 2, further comprising the step of:

if a number of lost audio packets counted by the counter during the time specified by the adjustment timer and is under a predetermined threshold, terminating the lost audio packet recovery process.

6. The method as recited in claim 2, further comprising steps, performed by the receiving station, of:

sending a request to the sending station for encoder/decoder parameter changes;

allocating a queue in memory to receive both a newly transmitted audio packet and a re-transmitted audio packet that are transmitted together;

receiving both the newly transmitted audio packet and the re-transmitted audio packet;

stopping the lost audio packet process; and

resetting a decoder to accept a plurality of new audio data packets transmitted by the sending station.

7. The method as recited in claim 6, further comprising the step of:

if the re-transmitted audio packet is a duplicate of a packet that has already been received, discarding the re-transmitted audio packet.

8. The method as recited in claim 6, further comprising the step of:

if the re-transmitted audio packet is too old and not useable with the current set of received audio packets, discarding the re-transmitted audio packet.

9. The method as recited in claim 6, further comprising the step of:

if the re-transmitted audio packet is one that was lost, reinserting the re-transmitted audio packet in an appropriate position among the other audio packets to produce an acceptable audio signal.

10. An apparatus for recovering lost packets representing audio data in a network environment, comprising:

a sender/receiver unit including:

a receiver; and

means for sending data, the sender/receiver unit communicating with a remote sender/receiver unit in the network, the receiver causing the sender/receiver unit to receive a new packet from the remote sender/receiver unit, wherein the packet contains a packet index, wherein the packet index is the numerical value that identifies the order of transmission for the packet; the receiver being further configured to perform steps, including:

storing a packet index of a last packet previously received in memory,

comparing the stored packet index and the packet index of the new packet to yield a difference;

starting, if the difference is greater than one, an adjustment timer and a counter to determine the number of lost audio packets for a specific period of time;

initiating a lost audio packet recovery procedure, if the adjustment timer and the counter show that the number of lost packets counted by the counter exceeds a predetermined threshold for the number of lost audio packets for the time specified by the adjustment timer; and

causing the sender/receiver unit to forward lost audio packet recovery information to the remote sender/receiver unit as part of the lost audio packet recovery procedure.

11. The apparatus as recited in claim 10, wherein the receiver considers the newly received audio packet out-of-sequence and discards the newly received packet, if the difference exceeds one half the value of the span of possible packet index values.

12. The apparatus as recited in claim 10, wherein the receiver terminates the lost audio packet recovery process if the adjustment timer reaches the specified time and the count is under a predetermined threshold for a predetermined period of time.

13. The apparatus as recited in claim 10, wherein the receiver terminates the lost audio packet recovery process, if the number of lost audio packets counted by the counter during the time specified by the adjustment timer is under a predetermined threshold.

14. The apparatus as recited in claim 10, in which the receiver is further configured to perform steps, including:

causing the sender/receiver unit to send a request to the remote sender/receiver unit for encoder/decoder parameters changes;

allocating a queue in memory to receive both a newly transmitted audio packet and a retransmitted audio packet that are transmitted together;

causing the sender/receiver unit to receive both the newly transmitted audio packet and the re-transmitted audio packet;

stopping the lost audio packet recovery process; and

resetting a decoder to accept a plurality of new audio data packets transmitted by the remote sender/receiver unit.

15. The apparatus as recited in claim 14, wherein the receiver discards the re-transmitted audio packet, if the re-transmitted audio packet is a duplicate of a packet that has already been received.

16. The apparatus as recited in claim 14, wherein the receiver discards the re-transmitted audio packet, if the re-transmitted audio packet is too old and not useable with the current set of received audio packets.

17. The apparatus as recited in claim 14, wherein receiver inserts the re-transmitted audio packet in an appropriate position among the other audio packets to produce an acceptable audio signal, if the re-transmitted audio packet is one that was previously lost.

18. A system for recovering lost packets representing audio data in a network environment, comprising:

first and second sender/receiver units communicating with each other and being configured for both transmitting and receiving packets representing audio data, each packet containing a index to uniquely identify the packet;

means for causing the first sender/receiver unit to start a counter and an adjustment timer to determine a number of lost packets for a specified time period, if a difference between the index of a new packet and the index of a last previous packet is greater than one; and

means for causing the first sender/receiver unit to initiate a lost packet recovery procedure when the number of lost packets for the specified period exceeds a

predetermined threshold, the lost packet recovery procedure includes sending a lost packet recovery information;

means for causing the second sender/receiver unit to receive the lost packet recovery information;

means for causing the second sender/receiver unit to provide a compression method suitably selected for recovery of the lost packets;

means for causing the first sender/receiver unit to prepare for recovering any lost packets, a lost packet being resent in tandem with a newly sent packet; and

means for causing both of the first and second sender/receiver units to stop the lost packet recovery procedure when the number of lost packets is under a specified threshold.

19. A method for recovering lost packets representing audio data in a network environment comprising:

receiving, by a receiver, packets representing audio data, each packet containing a index to uniquely identify the packet;

counting, in the receiver, a number of lost packets for a specified time period if a difference between the index of a previously received packet and the index of a currently received packet is greater than one;

initiating, by the receiver, a lost packet recovery procedure when the number of lost packets for the specified period exceeds a predetermined threshold, the lost packet recovery procedure includes sending, from the receiver to a sender, a lost packet recovery information;

providing, in the sender, a compression method suitably selected for recovery of the lost packets;

preparing, in the receiver, to recover any lost packets resent by the sender, for each lost packet resent by the sender the lost packet is sent in tandem with a new packet; and

sending from the receiver to the sender a stop recovery message when the number of lost packets is under a specified threshold.

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L14: Entry 3 of 9 File: USPT Jan 22, 2002

US-PAT-NO: 6341129
DOCUMENT-IDENTIFIER: US 6341129 B1

TITLE: TCP resegmentation

DATE-ISSUED: January 22, 2002

INVENTOR-INFORMATION:

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Hathaway; Wayne	Sunnyvale	CA		

ASSIGNEE-INFORMATION:

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APPL-NO: 9/ 055031 [PALM]
DATE FILED: April 3, 1998

INT-CL: [7] H04 L 12/28, H04 J 3/24

US-CL-ISSUED: 370/354; 370/389, 370/471, 370/474
US-CL-CURRENT: 370/354; 370/389, 370/471, 370/474

FIELD-OF-SEARCH: 370/216, 370/217, 370/218, 370/352, 370/353, 370/354, 370/397, 370/389, 370/392, 370/401, 370/465, 370/466, 370/469, 370/474, 370/470, 370/471, 370/496, 370/476, 709/232, 709/239

PRIOR-ART-DISCLOSED:

U.S. PATENT DOCUMENTS

Search Selected | Search ALL

	PAT-NO	ISSUE-DATE	PATENTEE-NAME	US-CL
☐	5065398	November 1991	Takashima	370/230
☐	5430727	July 1995	Callon	370/401
☐	5568477	October 1996	Galand et al.	370/229
☐	5715250	February 1998	Watanabe	370/395
☐	5793983	August 1998	Albert et al.	709/239
☐	5805822	September 1998	Long et al.	709/232
☐	5818842	October 1998	Burwell et al.	370/397
☐	5889777	March 1999	Miyao et al.	370/466
☐	6038231	March 2000	Dolby et al.	370/474

OTHER PUBLICATIONS

William Stallings, "Data and Computer Communications" pp. 136-139, 1985.*
Internet Protocol, Darpa Internet Program, Protocol Specification, Sep. 1981.
J.Postel, Internet Control Message Protocol, Darpa Internet Program, Protocol Specfication, Sep. 1981.
Transmission Control Protocol, Darpa Internet Program, Protocol Specification, Sep. 1981.
Internet Standards Archive, Oct. 1989.
V,. Jacobsen, TCP Extensions for Long-Delay Paths, Oct. 1989.
J. Postel, The TCP Maximum Segment Size and Related Topics, Nov. 1983.

ART-UNIT: 2662
PRIMARY-EXAMINER: Hsu; Alpus H.
ASSISTANT-EXAMINER: Qureshi; Afsar M.
ATTY-AGENT-FIRM: Glenn; Michael A.

ABSTRACT:

A resegmentation entity implements a TCP resegmentation technique wherein a receiving host receives packets that appear as if it they have been transmitted specifically for the receiving host's MTU. The receiving host does not require the buffering and CPU utilization necessary for IP reassembly. Thus, the receiving host has a lower latency when receiving IP datagrams that contain resegmented TCP segments than it would if it needed to re-assemble an IP datagram from fragments before it could process the TCP segment. Further, the sending host transmits TCP segments at its largest MTU, without regard to the receiving station's MTU, knowing that the intermediate routing entity insures that TCP resegmentation occurs. In the event that an IP datagram containing a resegmented TCP segment is lost, the sending host only has to retransmit the actual TCP data that was lost, and not the complete TCP segment.

24 Claims, 6 Drawing figures

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L14: Entry 1 of 9 File: USPT May 21, 2002

US-PAT-NO: 6392993
DOCUMENT-IDENTIFIER: US 6392993 B1

TITLE: Method and computer program product for efficiently and reliably sending small data messages from a sending system to a large number of receiving systems

DATE-ISSUED: May 21, 2002

INVENTOR-INFORMATION:

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Meizlik; Robert Steven	Newcastle	WA		

ASSIGNEE-INFORMATION:

NAME	CITY	STATE	ZIP CODE	COUNTRY	TYPE CODE
Microsoft Corporation	Redmond	WA			02

APPL-NO: 9/ 106403 [PALM]
DATE FILED: June 29, 1998

PARENT-CASE:
RELATED APPLICATIONS This application is related to U.S. patent application Ser. No. 09/106,531, entitled "Method and Computer Program Product for Efficiently and Reliably Sending Small Data Messages From a Sending System to a Large Number of Receiving Systems," filed in the names of Keith Hamilton and Robert Meizlik now U.S. Pat. No. 6,112,323, and to U.S. patent application Ser. No. 09/106,400, entitled "Method and Computer Program Product for Efficiently and Reliably Sending Small Data Messages From a Sending System to a Large Number of Receiving Systems," filed in the names of Keith Hamilton and Robert Meizlik, both of which were filed on the same date as the present application.

INT-CL: [7] H04 J 3/26

US-CL-ISSUED: 370/230; 370/474, 714/748
US-CL-CURRENT: 370/230; 370/474, 714/748

FIELD-OF-SEARCH: 370/229-232, 370/235, 370/236, 370/390, 370/474, 714/748, 714/749

PRIOR-ART-DISCLOSED:

U.S. PATENT DOCUMENTS

Search Selected | Search ALL

	PAT-NO	ISSUE-DATE	PATENTEE-NAME	US-CL
☐	5627970	May 1997	Keshav	395/200.13
☐	6097697	August 2000	Yao et al.	370/230
☐	6112323	August 2000	Meizlik et al.	714/748

OTHER PUBLICATIONS

RFC 768; J. Postel; ISI; Aug. 28, 1980; User Datagram Header Format; (pp. 1-3).
RFC 792; Message Formats; Sep. 1981.
RFC 1112; Deering, S.; Host Extensions for IP Multicasting; (pp. 1-16); Stanford University; Aug. 1989.
RFC 1122; Postel. J; Network Working Group--Internet Control Message Protocol:DARPA Internet Program Protocol Specification; Sep. 1981, (pp. 1-43).
Chapter 3: Specification; Jan. 1980; Internet Protocol; (pp. 11-41).
Muuss, Mike; Code derived from Software Contributed to Berkeley by Mike Muuss; The Regents of the University of California; Copyright .COPYRGT. 1989, 1993 (pp. 1-37).

ART-UNIT: 2663
PRIMARY-EXAMINER: Marcelo; Melvin
ATTY-AGENT-FIRM: Workman, Nydegger, Seeley

ABSTRACT:

In a network with a sending system networked to at least one receiving system, it is sometimes desirable to transfer relatively short messages between the sending system and one or more receiving systems in a highly reliable yet highly efficient manner. The present invention defines two short message protocols, one of which relies on a statistical model and the other of which uses positive acknowledgement to track receipt of transmitted packets by intended recipient. The statistical reliability mode is based on the observation that for each packet in a message that is transmitted, the probability that at least one packet of the message is received by a given system increases. Thus, in the statistical reliability mode messages are divided into a guaranteed minimum number of packets, with additional packets being added if the message length is insufficient to fill the minimum number of packets. The positive reliability mode of the present invention periodically sets an acknowledgement flag in the packets transmitted for a message. Receiving systems send an acknowledgement in response to receipt of that packet. The sending system tracks receipt of acknowledgements by intended recipient and retransmits any unacknowledged packets so as to positively assure the packets are received. Receiving systems send negative acknowledgements to request retransmission of missing packets. Negative acknowledgement suppression is implemented at both the sender and receiver to prevent a flood of negative acknowledgements from overwhelming the network. Packets are transmitted by the sending system at a transmission rate selected to avoid any adverse impact on the packet loss rate of the network.

29 Claims, 14 Drawing figures